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| EXAMINER | |
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| WEST, JEFFREY R | |
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**Please find below and/or attached an Office communication concerning this application or proceeding.**

The time period for reply, if any, is set in the attached communication.

| Office Action Summary | Application No. | Applicant(s) | |
|---|---|---|---|
| | 10/758,176 | REYNOLDS ET AL. | |
| | Examiner | Art Unit | |
| | Jeffrey R. West | 2857 | |

*-- The MAILING DATE of this communication appears on the cover sheet with the correspondence address --*

**Period for Reply**

A SHORTENED STATUTORY PERIOD FOR REPLY IS SET TO EXPIRE *3* MONTH(S) OR THIRTY (30) DAYS, WHICHEVER IS LONGER, FROM THE MAILING DATE OF THIS COMMUNICATION.

- Extensions of time may be available under the provisions of 37 CFR 1.136(a). In no event, however, may a reply be timely filed after SIX (6) MONTHS from the mailing date of this communication.
- If NO period for reply is specified above, the maximum statutory period will apply and will expire SIX (6) MONTHS from the mailing date of this communication.
- Failure to reply within the set or extended period for reply will, by statute, cause the application to become ABANDONED (35 U.S.C. § 133). Any reply received by the Office later than three months after the mailing date of this communication, even if timely filed, may reduce any earned patent term adjustment. See 37 CFR 1.704(b).

**Status**

1)☒ Responsive to communication(s) filed on *29 December 2008*.

2a)☒ This action is **FINAL**. 2b)☐ This action is non-final.

3)☐ Since this application is in condition for allowance except for formal matters, prosecution as to the merits is closed in accordance with the practice under *Ex parte Quayle*, 1935 C.D. 11, 453 O.G. 213.

**Disposition of Claims**

4)☒ Claim(s) *1-5 and 9* is/are pending in the application.

4a) Of the above claim(s) _____ is/are withdrawn from consideration.

5)☐ Claim(s) _____ is/are allowed.

6)☒ Claim(s) *1-5 and 9* is/are rejected.

7)☐ Claim(s) _____ is/are objected to.

8)☐ Claim(s) _____ are subject to restriction and/or election requirement.

**Application Papers**

9)☐ The specification is objected to by the Examiner.

10)☒ The drawing(s) filed on *20 March 2006* is/are: a)☒ accepted or b)☐ objected to by the Examiner.

Applicant may not request that any objection to the drawing(s) be held in abeyance. See 37 CFR 1.85(a).

Replacement drawing sheet(s) including the correction is required if the drawing(s) is objected to. See 37 CFR 1.121(d).

11)☐ The oath or declaration is objected to by the Examiner. Note the attached Office Action or form PTO-152.

**Priority under 35 U.S.C. § 119**

12)☒ Acknowledgment is made of a claim for foreign priority under 35 U.S.C. § 119(a)-(d) or (f).

a)☒ All b)☐ Some * c)☐ None of:

1.☒ Certified copies of the priority documents have been received.

2.☐ Certified copies of the priority documents have been received in Application No. _____.

3.☐ Copies of the certified copies of the priority documents have been received in this National Stage application from the International Bureau (PCT Rule 17.2(a)).

* See the attached detailed Office action for a list of the certified copies not received.

**Attachment(s)**

1) ☒ Notice of References Cited (PTO-892)

2) ☐ Notice of Draftsperson's Patent Drawing Review (PTO-948)

3) ☐ Information Disclosure Statement(s) (PTO/SB/08) Paper No(s)/Mail Date _____.

4) ☐ Interview Summary (PTO-413) Paper No(s)/Mail Date. _____ .

5) ☐ Notice of Informal Patent Application

6) ☐ Other: _____.

## DETAILED ACTION

1. This application currently names joint inventors. In considering patentability of the claims under 35 U.S.C. 103(a), the examiner presumes that the subject matter of the various claims was commonly owned at the time any inventions covered therein were made absent any evidence to the contrary. Applicant is advised of the obligation under 37 CFR 1.56 to point out the inventor and invention dates of each claim that was not commonly owned at the time a later invention was made in order for the examiner to consider the applicability of 35 U.S.C. 103(c) and potential 35 U.S.C. 102(e), (f) or (g) prior art under 35 U.S.C. 103(a).

### *Claim Rejections - 35 USC § 103*

2. The following is a quotation of 35 U.S.C. 103(a) which forms the basis for all obviousness rejections set forth in this Office action:

> (a) A patent may not be obtained though the invention is not identically disclosed or described as set forth in section 102 of this title, if the differences between the subject matter sought to be patented and the prior art are such that the subject matter as a whole would have been obvious at the time the invention was made to a person having ordinary skill in the art to which said subject matter pertains. Patentability shall not be negatived by the manner in which the invention was made.

3. Claims 1 and 9 are rejected under 35 U.S.C. 103(a) as being unpatentable over Dechjaroen, “Performance Evaluation of Voice Over Internet Protocol”, in view of U.S. Patent No. 6,665,317 to Scott and further in view of Rix et al., "Non-intrusive monitoring of speech quality in voice over IP networks".

With respect to claim 1, Dechjaroen discloses a method of assessing speech quality transmitted via a packet based telecommunications network comprising the steps of: storing a sequence of intercepted packets associated with a call (page 5,

lines 6-8), each packet containing speech data (page 7, lines 1-4), and an indication of a transmission time of said intercepted packet (page 43, lines 15-17); storing with each intercepted packet an indication of an intercept time of said packet (page 48, lines 21-24); extracting a set of parameters from said sequence of intercepted packets (page 49, lines 1-10); wherein the extracting step comprises the sub steps of: generating a jitter parameter for each packet of said sequence of stored packets in dependence upon a difference between the transmission time of a stored packet and the transmission time of a preceding stored packet of the sequence and a difference between the intercept time of said stored packet and the intercept time of said preceding stored packet (page 49, lines 1-6 and Figure 29); generating a long term average jitter parameter (lt_jitter) for said stored packet in dependence upon the value of said jitter parameter (jitter) for said stored packet, the value of said jitter parameter for any preceding stored packets, and a predetermined adaptation rate (P) according to the equation: lt_jitter = (lt_jitter * P) + (abs (jitter) * (1 – P) (i.e. J = J+(|D(i – 1, i) | – J) / 16) (page 49, line 7 to page 50, line 3).

With respect to claim 9, Dechjaroen discloses an apparatus for assessing speech quality transmitted via a packet based telecommunications network comprising: means for storing a sequence of intercepted packets associated with a call (page 5, lines 6-8), each packet containing speech data (page 7, lines 1-4), and an indication of a transmission time of said packet (page 43, lines 15-17); means for storing with each intercepted packet an indication of an intercept time of said intercepted packet (page 48, lines 21-24); means for extracting a set of parameters from said sequence

of intercepted packets (page 49, lines 1-10); wherein the means for extracting further comprises: means for generating a jitter parameter for each intercepted packet of said sequence of stored intercepted packets in dependence upon a difference between the transmission time of a stored intercepted packet and the transmission time of a preceding stored packet of the sequence and a difference between the intercept time of said stored intercepted packet and the intercept time of said preceding stored intercepted packet (page 49, lines 1-6 and Figure 29); means for generating a long term average jitter parameter (lt_jitter) for said stored packet in dependence upon the value of said jitter parameter (jitter) for said stored intercepted packet, the value of said jitter parameter for any preceding stored intercepted packets, and a predetermined adaptation rate (P) according to the equation: lt_jitter = (lt_jitter * P) + (abs (jitter) * (1 – P) (i.e. J = J+(|D(i – 1, i) | – J) / 16) (page 49, line 7 to page 50, line 3).

As noted above, the invention of Dechjaroen teaches many of the features of the claimed invention and while the invention of Dechjaroen does teach extracting a set of jitter parameters including inter-packet jitter and long-term average jitter, Dechjaroen does not explicitly include means for determining a differential jitter parameter.

Scott teaches a method, system, and computer program product for managing jitter of packets across a VoIP system (column 1, line 65 to column 2, line 2) that calculates a long term jitter parameter in dependence upon a value of jitter for a stored packet and a value of jitter for any preceding stored packets (column 5, lines

22-23 and 41-46) and a differential jitter (i.e. jitter variance) in dependence upon the jitter parameter and the long term jitter parameter (column 5, lines 22-25).

It would have been obvious to one having ordinary skill in the art to modify the invention of Dechjaroen to explicitly include means for determining a differential jitter parameter of the extracted parameters, as taught by Scott, because, as suggested by Scott, the combination would have improved the speech quality analysis of Dechjaroen by determining a more complete group of jitter parameters including a jitter variation which would provide an indication as to the changes in the size of a packet from the start to destination thereby allowing the user to monitor such a size change for determining a point of insufficient quality and/or times of congestion (column 3, line 66 to column 4, line 4).

As noted above, the invention of Dechjaroen and Scott teaches many of the features of the claimed invention and while the invention of Dechjaroen and Scott does teach extracting a set of jitter parameters including inter-packet jitter, long-term average jitter, and differential jitter, the combination does not specify determining an estimated mean opinion score in dependence upon said set of parameters.

Rix teaches non-intrusive monitoring of speech quality in VoIP networks comprising storing a sequence of intercepted packets, containing speech data, associated with a call (page 4, lines 23-24 and “Capture” in Figure 3), extracting a set of jitter parameters from said sequence of intercepted packets (page 5, line 4 and “Extract Params” in Figure 3), and generating, and inherently storing on a medium for user-visualization/analysis, an estimated mean opinion score in

dependence upon said set of parameters (page 5, lines 5-6 and “Predict MOS” in Figure 3).

It would have been obvious to one having ordinary skill in the art to modify the invention of Dechjaroen and Scott to specify determining an estimated mean opinion score in dependence upon said set of parameters, as taught by Rix, because one having ordinary skill in the art would recognize that a mean opinion score is a common meter of VoIP quality and, as suggested by Rix, the combination would have improved the speech quality analysis of Dechjaroen and Scott by automatically determining a common index of speech quality in order to track quality over time, using the determined jitter variation/differential jitter of Dechjaroen and Scott, to provide a real-time measure of call quality (page 5, lines 15-28).

4. Claims 2-5 are rejected under 35 U.S.C. 103(a) as being unpatentable over Dechjaroen in view of Scott and Rix et al. and further in view of U.S. Patent Application Publication No. 2003/0018450 to Carley.

As noted above, Dechjaroen in combination with Scott and Rix teaches many of the features of the claimed invention and while the invention of Dechjaroen, Scott, and Rix does teach extracting a set of parameters from a sequence of packets including a jitter parameter, long term average jitter parameter, and differential jitter parameter, the combination does not specifically include determining a maximum or variance value of the measured parameters and a subsequent average of the maximum and/or variance value.

Carley teaches a system and method for providing composite variance analysis for network operation of a packet based network (0002, lines 1-9 and 0017, line 1 to 0024, line 3) comprising means for extracting and storing a jitter parameter performance metric for a sequence of packets (0041, lines 1-23) determining a variance statistic for the performance metric (0045, lines 1-8) and determining a subsequent standard deviation of the determined variance statistic (0047, line 4 to 0048, line 7), wherein the variance statistic includes a plurality of maximum values and standard deviations of sub-sequences of the performance metric (0068, lines 11-19). Therefore, Carley teaches determining both a maximum of the performance metric followed by a standard deviation of the maximum as well as a standard deviation of the performance metric followed by a subsequent standard deviation. It is further considered inherent that in order to determine each standard deviation, an average and variance must first be determined (see for example, Internet Glossary of Statistical Terms, "Variance" and "Standard Deviation").

It would have been obvious to one having ordinary skill in the art to modify the invention of Dechjaroen, Scott, and Rix to include determining a maximum and variance value of the measured parameters and a subsequent average of the maximum and/or variance value, as taught by Carley, because the invention of Dechjaroen, Scott, and Rix does teach a method for assessing the quality of speech packets but provides no significant method for determining when a speech quality degrades below a desired level and the invention of Carley suggests that the combination would have improved the method of Dechjaroen, Scott, and Rix and by

allowing the user to determine the quality with greater detail by determining how the performance of a given network server is performing with respect to any desired performance metric over time as well as determine whether the performance of a network service at any particular time is outside of acceptable limits (0040, lines 1-28).

## ***Response to Arguments***

5. Applicant's arguments filed December 29, 2008, have been fully considered but they are not persuasive.

Applicant argues:

> Scott '317 discloses that it is known to generate jitter statistics including jitter, jitter variation, average jitter, average jitter variation and combinations thereof for the purpose of managing a jitter buffer size. Handling jitter comes at the expense of latency, and Scott '317 addresses the problem of providing optimal buffering so that jitter is handled effectively without resorting to excessive buffering. Accordingly, the applicant further acknowledges that Scott '317 discloses the use of jitter statistics for a purpose that differs from the purpose of assessing speech quality.
>
> The U.S. PTO continues to assert, unfairly and without basis in the opinion of the applicant, that it would be obvious to generate a differential jitter parameter in dependence upon the jitter parameter for a stored packet and the long term average jitter parameter, and using this parameter to assess speech quality.
>
> It is respectfully urged that simply because use of a particular parameter is known for other purposes does not mean that use of such a parameter would be obvious in the context of a method of assessing speech quality.

The Examiner disagrees with Applicant's assertion that Scott uses jitter statistics for a purpose that differs from the purpose of assessing speech quality and instead, the Examiner asserts that Scott explicitly ties jitter to speech quality:

Latency and jitter each impact communication differently. For example, if packets always arrived 50 milliseconds (ms) after being transmitted, then there would be a 50 ms latency and no jitter. In another example, however, if packet #1 arrived 100 ms after transmission, packet #2 arrived 50 ms after transmission, and packet #3 arrived 150 ms after transmission, there would be an average jitter of +/-33 ms. In voice over Internet protocol (VoIP) applications, jitter is more critical than latency. Jitter can cause a packet to arrive too late to be useful. The effect is that the packet may be delayed enough that the end user will hear a pause in the voice that is talking to them, which is very unnatural if it occurs during the middle of a word or sentence. (column 1, lines 21-33)

One shortfall of early VoIP systems was the poor quality of voice jittery voice) and the unacceptable latency caused by the fluctuating, and at times less than adequate bandwidth available through the Internet.

According to the present invention, jitter buffer managing is used to resolve the quality of voice over the unpredictable and some time limited bandwidth of the Internet. This capability adjusts the size and contents of the jitter buffer, thus minimizing jitter. (column 3, lines 13-21)

Applicant argues:

The equation used in Dechjaroen is disclosed therein only in the context of adapting the buffer length in order to optimize the performance of the buffer, i.e., for an application which is similar to that disclosed in Scott '317. Nowhere does Dechjaroen (or any other of the cited references) disclose of suggest that such a long term average jitter should be used as a parameter in a method of assessing speech quality.

Furthermore, Dechjaroen does not disclose or suggest determining a difference between a jitter parameter and a long term average jitter for any purpose. Rix is relied on for alleged disclosure of estimating a mean opinion score, in which jitter parameters are used to determine a mean opinion score. Any such disclosure in Rix would not overcome the shortcomings of Dechjaroen and Scott '317 as attempted to be applied against claim 1.

Applicants respectfully submit that finding disclosure of a particular parameter (e.g., differential jitter parameter) does not render obvious use of that parameter in another method. Here, Scott '317 discloses that it is known to generate jitter statistics including jitter variation, average jitter and average jitter variation for the purposes of managing a jitter buffer size. Similarly, Dechjaroen is also concerned with optimizing the performance of the buffer. For example, the cited equation on page 49 that the Examiner relies on is used in the context of adapting the buffer length in order to optimize performance of the buffer, which is a similar application as shown in Scott '317.

First, the Examiner asserts that Applicant's argument that "Dechjaroen does not disclose or suggest determining a difference between a jitter parameter and a long term average jitter for any purpose" is not considered to be persuasive because the invention of Scott is relied upon for teaching a method, system, and computer program product for managing jitter of packets across a VoIP system (column 1, line 65 to column 2, line 2) that calculates a long term jitter parameter in dependence upon a value of jitter for a stored packet and a value of jitter for any preceding stored packets (column 5, lines 22-23 and 41-46) and a differential jitter (i.e. jitter variance) in dependence upon the jitter parameter and the long term jitter parameter (column 5, lines 22-25).

Second, while Applicant argues that Scott's teaching of a differential jitter is for optimizing buffer performance, the Examiner asserts that, as discussed above, Scott is concerned with determining differential jitter in order to improve speech quality.

Third, while Applicant argues that Dechjaroen is also concerned with optimizing buffer performance, the Examiner asserts that Dechjaroen is similarly concerned with using measured jitter parameters to assess speech quality, specifically:

> Voice over Internet Protocol (VoIP) was developed to emulate toll services with lower communication cost. In VoIP applications, voices are digitized and packetized into small blocks. These voice blocks are encapsulated in a sequence of voice packets using the Real-time Transport Protocol (RTP) and delivered by the User Datagram Protocol (UDP). To help VoIP applications deal with unpredictable network performance, the Real-time Transport Control Protocol (RTCP) is developed to monitor the performance of RTP packets and provide feedback to the VoIP applications. The feedback on packet delay, jitter, and loss

rate enables the applications to adapt to network conditions to maintain a certain level of voice quality. With this architecture, the quality of service of VoIP relies on the effectiveness of the RTCP network performance report mechanism. (abstract)

Real-time Transport Control Protocol (RTCP) serves as a control counterpart of the RTP operation. This protocol reports the data distribution quality periodically in the form of sender and receiver reports. The RTP source can also use RTCP to help its receiver synchronize audio and video input. (page 12, lines 5-8)

The information provided inside RTCP messages can be used to evaluate the performance of the associated real-time continuous media application because RTCP indirectly reports the quality of service in the network. Each report block is sent with the collective management information, such as the latest sequence number received, the number of missing packets, and jitter. However, RFC 1889 does not specify how to use these values. (page 19, lines 14-19)

Since VoIP is designed to emulate the toll services, the quality of packetized voice is the key concern. In the existing environment, the public networks cannot guarantee VoIP reliability and sound quality like the PSTN communication due to the limitation on network bandwidth. To determine the performance of VoIP, several factors should be considered. - specifically delay, jitter, packet loss, and echo. This chapter discusses these factors and the source of voice degradation. (page 25, lines 3-8)

The virtual speech is generated by computer using network programming in which the payload portion in voice packet can be any bit stream. The important contents - RTP, UDP, and IP header - carries network performance information, such as delay, jitter, and packet loss. (page 39, lines 23-26)

In order to measure the performance directly, voice packet is generated and transmitted on the real network or on dedicated channel simulator. Test may include a central office switch, gateway, and gatekeeper. Since voice packet can be manipulated at a source, it is quite flexible to derive the output from a header info. After evaluation, the analytical data collected at a receiver is compared to the source data. Finally, the performance parameters - such as delay, jitter, packet loss, and packet unorder - can be determined. (page 39, lines 7-13)

Applicant argues:

> Applicants have found that the recited long term average jitter parameter and the differential jitter parameter are particularly important parameters in determining and generating a mean opinion score relating to the voice quality of a VoIP call. None of the references alone or in combination, disclose any direct relationship between these elements and a mean opinion score. Rather, for example, Scott '317 discloses that an average jitter value and a differential jitter value are used as part of calculations to adjust the size of a jitter buffer. Scott '317 does not disclose or suggest that the average jitter value and the differential jitter value can or should be used as a rating from which a mean opinion score can be determined. The other references also fail to contain any such disclosure or suggestion.

The Examiner asserts that Dechjaroen explicitly discloses using the determined jitter parameters for determining a Mean Opinion Score, specifically:

> Measurement in real communication can also be used for objective test. It requires some computation on packet header contents. Delay, jitter, and packet loss can be determined from RTCP packet. Then all parameters can be converse to MOS by using E-model. (page 42, lines 9-12)
>
> In order to use the network parameters - such as delay, jitter, and packet loss – to tune the data networks, these objective numbers must be mapped to the subjective value such as MOS. The most acceptable conversion model called "E-model" is recommended in ITU G.107 [Ref 29]. It is used by NetIQ [Ref 28] for VoIP performance testing application. This model requires two mechanisms: calculating the R-Value and mapping to MOS. (page 52, lines 4-9)

Therefore, while Dechjaroen does teach extracting a set of jitter parameters including inter-packet jitter and long-term average jitter and using such parameters to determine an MOS, Dechjaroen does not explicitly include means for determining a differential jitter parameter for calculating the MOS.

Scott then teaches a method, system, and computer program product for managing jitter of packets across a VoIP system (column 1, line 65 to column 2, line

2) that calculates a long term jitter parameter in dependence upon a value of jitter for a stored packet and a value of jitter for any preceding stored packets (column 5, lines 22-23 and 41-46) and a differential jitter (i.e. jitter variance) in dependence upon the jitter parameter and the long term jitter parameter (column 5, lines 22-25) that would have improved the speech quality analysis of Dechjaroen by determining a more complete group of jitter parameters including a jitter variation which would provide an indication as to the changes in the size of a packet from the start to destination thereby allowing the user to monitor such a size change for determining a point of insufficient quality and/or times of congestion (column 3, line 66 to column 4, line 4).

Rix then teaches non-intrusive monitoring of speech quality in VoIP networks comprising storing a sequence of intercepted packets, containing speech data, associated with a call (page 4, lines 23-24 and “Capture” in Figure 3), extracting a set of jitter parameters from said sequence of intercepted packets (page 5, line 4 and “Extract Params” in Figure 3), and generating, and inherently storing on a medium for user-visualization/analysis, an estimated mean opinion score in dependence upon said set of parameters (page 5, lines 5-6 and “Predict MOS” in Figure 3).

As such, the Examiner maintains that the combination teaches determining a differential jitter parameter and generating an estimated Mean Opinion Score from the determined differential jitter parameter.

## *Conclusion*

6. The prior art made of record and not relied upon is considered pertinent to Applicant's disclosure:

Demichelis et al., "IP Packet Delay Variation Metric for IP Performance Metrics (IPPM) teaches methods for determining metrics for variation in delay of packets across Internet paths.

Internet Glossary of Statistical Terms, “Variance” and “Standard Deviation” teaches the definitions for “Variance” and “Standard Deviation” as well as that in order to calculate the variance, a mean/average must first be determined, as well as that in order to calculate the standard deviation, a variance must first be determined.

Schulzrinne teaches a method and system for determining jitter measurements of packets across IP in audio/voice applications (“2. RTP Use Scenarios” and “2.1 Simple Multicast Audio Conference”) including means for generating a long term average jitter parameter (lt_jitter / J) for a stored packet in dependence upon a value of a jitter parameter (jitter / D (i –1, i)) for said stored packet, a value of said jitter parameter for any preceding stored packets, and a predetermined adaptation rate (P / 15/16) according to the equation lt_jitter = (lt_jitter * P) + (abs (jitter) * (1 – P) (i.e. J = J+(|D(i – 1, i) | – J) / 16) (“6.3.1 SR: Sender report RTCP packet”).

U.S. Patent Application Publication No. 2003/0086425 to Bearden teaches network traffic generation and monitoring systems and methods for their use in testing frameworks for determining suitability of a network for target applications, such as VoIP network applications (0006, lines 1-10), comprising means for

extracting a set of speech quality parameters, including jitter, and, generating an estimated mean opinion score in dependence upon the set of speech quality parameters (0085, lines 1-13) and storing the estimated mean opinion score on a computer-readable medium accessible by a user for visualization and analysis (0259, lines 1-19).

U.S. Patent Application Publication No. 2004/0071170 to Fukuda teaches a communication system, transmission terminal, and reception terminal

U.S. Patent No. 6,868,094 to Bordonaro et al. teaches a method and apparatus for measuring network data packet delay, jitter and loss.

7. **THIS ACTION IS MADE FINAL.** Applicant is reminded of the extension of time policy as set forth in 37 CFR 1.136(a).

A shortened statutory period for reply to this final action is set to expire THREE MONTHS from the mailing date of this action. In the event a first reply is filed within TWO MONTHS of the mailing date of this final action and the advisory action is not mailed until after the end of the THREE-MONTH shortened statutory period, then the shortened statutory period will expire on the date the advisory action is mailed, and any extension fee pursuant to 37 CFR 1.136(a) will be calculated from the mailing date of the advisory action. In no event, however, will the statutory period for reply expire later than SIX MONTHS from the mailing date of this final action.

8. Any inquiry concerning this communication or earlier communications from the examiner should be directed to JEFFREY R. WEST whose telephone number is (571)272-2226. The examiner can normally be reached on Monday through Friday, 8:30-5:00.

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, Eliseo Ramos-Feliciano can be reached on (571)272-7925. The fax phone number for the organization where this application or proceeding is assigned is 571-273-8300.

Information regarding the status of an application may be obtained from the Patent Application Information Retrieval (PAIR) system. Status information for published applications may be obtained from either Private PAIR or Public PAIR. Status information for unpublished applications is available through Private PAIR only. For more information about the PAIR system, see http://pair-direct.uspto.gov. Should you have questions on access to the Private PAIR system, contact the Electronic Business Center (EBC) at 866-217-9197 (toll-free). If you would like assistance from a USPTO Customer Service Representative or access to the automated information system, call 800-786-9199 (IN USA OR CANADA) or 571-272-1000.

/Jeffrey R. West/
Primary Examiner, Art Unit 2857

April 10, 2009